REMARKS

Status of the Application:

Claims 1–17 are the claims of record of the application. Claims 1–17 have been rejected.

Amendment to the Claims:

Applicants have cancelled claims 2 and 10, and added these features in claims 1 and 9, respectively. Applicants also have amended the independent claims to more clearly identify the invention, and to more clearly distinguish the invention of the cited art.

Applicants have also added new claims 17 through 20 that relate to claim 17 in the same way that claims 3 to 5 relate to claim 1 (as amended).

Claim Rejections -35 USC § 112 First Parag. (Nonenablement)

In paragraph 2 of the office action, claims 1–17 were rejected under 35 U.S.C. 112, first paragraph, as failing to comply with the enablement requirement. The Examiner asserted that the claim(s) contain subject matter which was not described in the specification in such a way as to enable one skilled in the art to which it pertains, or with which it is most nearly connected, to make and/or use the invention. The examiner asserted, in particular, that the specification fails to clearly disclose how filter coefficients for each filter stage from an associated group of sample points are determined out of the first plurality of sample points. The examiner asserts that there is insufficient disclosure to enable a person of ordinary skill in the art to figure out how to derive the transform matrices (62_0 to 62_n) in order to determine the filter coefficients for each filter stage.

The Applicant respectfully disagree that the specification lacks enablement that would enable one of ordinary skill in the art to practice the invention, and in particular, to determine the filter coefficients.

One of ordinary skill in the art would be familiar with digital signal processing and filter design, as this is the subject matter of the invention. For example, it is assumed that one would have sufficient skill to read the Keele article cited by the Examiner. Keele in

turn cites some pretty standard texts on digital signal processing, such as A. V. Oppenheiin and R. W. Schafer, "Discrete-Time Signal Processing," Prentice Hall, New Jersey (1989) (Keele's reference [I]), A. Antoniou, "Digital Filters, Analysis, Design, and Applications," McGraw-Hill, New York (1993) (Keele's reference [6]), and R. A. Roberts, C. L. Mullis, "Digital Signal Processing," Addison-Wesley, Reading, Norwood, MA (1987) (Keele's reference [7]).

As admitted by the Examiner, one embodiment teaches using the "transform matrices 62_0 to 62_n" in order to compute the filter coefficients. Clearly the method relies on the use of coefficients within these transform matrices that achieve the desired outcome, and thus, Applicants reasonably interpret the Examiner's rejection as asserting that there is insufficient disclosure to determine these coefficients.

How to determine these coefficients is described in the specification in parags [0050] to [0051] to include "a pseudo inverse filter response characteristic analysis". One of ordinary skill in the art, e.g., one able to read and understand the present application and the prior art cited by Keele would certainly have a level of skill to understand this term.

Nevertheless, the following makes it even clearer that the specification contains sufficient information:

Applicant's Figure 3 and the discussion thereof shows how each "transform" (e.g. 62.1) converts data points from the time-scale (32) to the set of filter taps (60.1). In effect, this diagram illustrates the process by which each FIR filter tap is determined by a mapping from the desired impulse response.

Applicant's Figure 5 and the discussion thereof shows plots of the impulse responses of each filter tap. It would be well known to one skilled in the art that these plots show how each filter tap generates a corresponding time signature, and that this is equivalent to a "reverse" process of the transforms in Figure 3. For example, the first filter tap (one of the taps in the group 60.1) has a characteristic impulse response that could be plotted on the time-line (32) in Figure 3, and hence this mapping from filter taps to

impulse responses is a mapping that goes in the opposite direction to the downward-directed arrows in Figure 3.

Of course the concept of an impulse response is well known to anyone of ordinary skill in the art of digital signal processing, and is explained in Figure 5 in terms of "impulse responses, corresponding to each of the FIR filter taps". Furthermore, the specification shows, e.g., in Figure 3, a mapping (a transform) from impulse response to filter taps, and, in Figure 5, shows the mapping from filter taps to impulse response. It turns out that, unsurprisingly, one can make a reasonable attempt to create the coefficients for the transforms (62.1 to 62.n) of Figure 3 by "inverting" the impulse response mappings shown in Figure 5.

It would be known to one of ordinary skill in the art that an "ideal" inverse mapping is not possible, and indeed, such an ideal inverse is not discussed in the specification. One of ordinary skill in the art would be familiar with how to achieve an approximation to the ideal. This is a least-mean-squares best fit approximation to the inverse mapping, and comes close to achieving the same result as the ideal inverse mapping. To anyone of ordinary skill in the art, this kind of "least-mean-squares best fit approximation to the inverse" is well known as the "pseudo-inverse." Indeed, use of the pseudo-inverse is what is suggested in the specification. The pseudo-inverse is also known as the generalized inverse. While whole textbooks have been written just on this inverse, and certainly, anyone familiar with signal processing would know that the pseudo-inverse provides an inverse in the least squares sense. See one of many basic textbook on matrix operations, e.g., Golub, G. H. & Van Loan, C. F. (1996) "Matrix Computation" (Johns Hopkins Univ. Press, Baltimore), 3rd Ed., Strang, G. (1976, 1988) "Linear Algebra and its Applications," G. Strang. *Linear algebra and its applications*. (New York, Academic Press, 2nd Ed. 1976, or Harcourt Brace Jovanovich College Publishers, 3rd Ed,1988). See also, Campbell, S. L. and Meyer, C. D. Jr. Generalized Inverses of Linear Transformations. New York: Dover, 1991. Rao, C. R. and Mitra, S. K. Generalized Inverse of Matrices and Its Applications. New York: Wiley, 1971. For simple Web-based descriptions, see for example, "Pseodo-invenrse, from Wikipedia, available at

en.wikipedia.org/wiki/Pseudo_inverse retrieved on Aug. 22, 2007. See also Weisstein, Eric W. "Moore-Penrose Matrix Inverse." From MathWorld--A Wolfram Web Resource, available at mathworld.wolfram.com/Moore-PenroseMatrixInverse.html retrieved on Aug 22, 2007, 2007. The material in these Websites is pretty basic, and would be known to one of ordinary skill in the art of filter design and digital signal processing.

The Examiner has therefore not made a prima-facie case of non-enablement for rejection under 35 USC 112, first paragraph. Reversal of the rejection is therefore respectfully requested.

Claim Rejections -35 USC § 102

In paragraph 4 of the office action, claims 1 and 17 were rejected under 35 U.S.C. 102(b) as being anticipated by Keele, Jr. "Log Sampling in Time and Frequency: Preliminary Theory and Application," hereinafter referrer to as "Keele."

In common with claims 1 and 17 of the presnt invention, Keele describes signal processing on samples that are scaled on logarithmic time and frequency scales. Keele is basically a theoretical paper that gives the theoretical underpinnings and desirability of such sampling.

Keele therefore computes filter coefficients for a log scale filter. These are Keele's a_1, $a_{2,,}$ a_3, etc.

As the examiner correctly points out, Keele in FIG. 22(a) and (b), Keele proposes a multirate structure for implementing a filter based on filter coefficients that are on a logarithmic scale.

Keele's design uses a pre-designed filter whose characteristics are not a fnctionof the desired response, and in addition, one log-scale coefficient per stage. Suppose, as an example, the decimation factor R is 2. This means that Keele's design has as many decimation stages as there are log-scale coefficients. As an example, suppose there are 13 log-scale coefficients. This means one would need 13 decimation stages, 13 interpolation

filters, going to 13 octave resolution. One of ordinary skill in the art would recognize that such a filter is not practical.

In Keele, more resolution is be achievable by using a non-integer decimation value, a lavue of R less than two, e.g., close to 1. The resulting structure is a non-conventional structure that has non-integer decimation.

The present invention claims a structure tat uses integer valued decimation, e.g., by a factor of 2, to be able to use more a conventional and easily implemented filter structure. The independent claims have been amended to more clearly bring out this feature.

One property of the present invention is, instead of using a single log-scale coefficient per muti-rate stage, to use in each stage a FIR filter, over a constant sample interval, that has more taps, the filter taps determined from more than one log-scale coefficient. The result is that, taking a decimation rate of two as an example, more accuracy can be achieved than the structure of Keele for the same number of filter stages. One feature of the present invention is determining these FIR filter coefficients from the log-scale coefficients (denoted a_1, a_2, etc. that Keele also computes). The filter taps are determined from more than one log scale coefficient. A feature of the invention describes how to compute these filter coefficients from more than one log-scale coefficient. Keele only shows a single log-scale coefficient per decimation stage.

In order to more clearly distinguish the present invention from Keene, Applicants have amended the independent claims to each recite that

> each successive filter stage in the multirate digital filter is for linearly spaced samples at a sampling rate that is decimated by an integer factor from the previous filter stage,

and further to include that

> at least one plurality of filter coefficients for a respective filter stage is determined from more than one log-scale sample point,

such that at least one of the successive filter stages in the multirate digital filter device has a filter response determined from more than one log-scale sample.

The claims are believed allowable in view of the explanation and amendment.

For these reasons, and in view of the above amendment, this application is now considered to be in condition for allowance and such action is earnestly solicited.

Conclusion

The Applicants believe all of Examiner's rejections have been overcome with respect to all remaining claims (as amended), and that the remaining claims are allowable. Action to that end is respectfully requested.

If the Examiner has any questions or comments that would advance the prosecution and allowance of this application, an email message to the undersigned at dov@inventek.com, or a telephone call to the undersigned at +1-510-547-3378 is requested.

Respectfully Submitted,

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